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RESEARCH DEPARTMENT



REPORT

**Pulse-code modulation for
high-quality sound signal distribution:
feasibility of multiplex system**

No. 1970/34

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FEASIBILITY OF MULTIPLEX SYSTEM**

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J.R. Chew, B.Sc.(Eng.), C.Eng., M.I.E.E.

D. Maurice
Head of Research Department

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PULSE-CODE MODULATION FOR HIGH-QUALITY SOUND SIGNAL DISTRIBUTION: FEASIBILITY OF MULTIPLEX SYSTEM

Summary

The feasibility has been investigated of applying p.c.m. to the distribution of stereophonic programmes of the highest broadcast quality in time-division multiplex. By this means all the programmes required for the various radio services could be carried over a single wide-band link.

Experimental equipment comprising a 14-bit coder and decoder, together with a 13-channel multiplexer and demultiplexer was developed for this purpose. The equipment functioned as predicted, and, while the amount of disturbance suffered by the sound signal when faults occur on the link could be reduced by further development, the practicability of the system as a whole is considered proven.

1. Introduction

In a report¹ issued in 1967, the advantages of pulse code modulation for the distribution of high-quality sound signals were discussed and the essential technical requirements considered. Work on this application of p.c.m. has since continued, and one of the possibilities originally envisaged, the incorporation of the television sound signal in the associated video waveform, has now been realised in practice;² this method of combining the two components of the programme is now known as 'Sound-in-Syncs'.

In the meantime, the problems of distributing a number of broadcast-quality sound signals in digital form on a common link have been investigated; for this purpose it was necessary to consider the functions of combining (multiplexing) and separating (demultiplexing) the individual signals at the sending and receiving terminals, of achieving and maintaining synchronism, and of minimising the subjective effect of occasional digit errors due to momentary interruptions or noise on the link.

Other investigations carried out in connection with this project indicated that to meet the needs of the V.H.F. radio service, and in particular the additional technical requirements imposed by stereophony, the number of bits per word, and possibly the sampling frequency, might have to be greater than originally estimated; experimental coding and decoding equipment of higher performance was therefore developed to meet the most exacting requirements that could be envisaged.

In the present report, the special features of a multiplex p.c.m. distribution system for broadcast programmes are discussed and recommendations are made for standardising essential parameters. The instrumentation of the experimental equipment developed to test the feasibility of the system proposed is described elsewhere.³

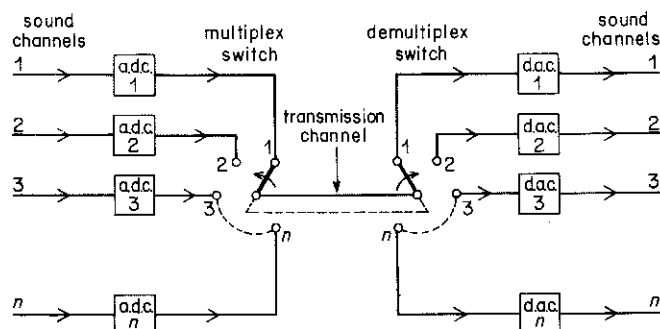


Fig. 1 - Simplified block diagram

2. Principles of multiplex p.c.m.

The principles of p.c.m. multiplex transmission are fairly well known but will be briefly recapitulated here. Fig. 1 shows a simplified block diagram of a p.c.m. multiplex system. Each analogue-to-digital converter (a.d.c. or — more succinctly — coder) samples the input signal from the sound channel at regular intervals and converts the sample voltage into an equivalent binary number. The multiplex switch reads the binary numbers from each a.d.c. in turn and sends them down the transmission channel. The demultiplex switch, which is synchronised to the multiplex switch, directs each number to the appropriate digital-to-analogue converter (d.a.c. or decoder) which reconverts the signal back to analogue form.

In practice the switches are electronic rather than mechanical. Each binary number read by the multiplex switch (multiplexer) results in the transmission of a succession of pulses, representing binary digits. The group of binary digits (bits) resulting from one sample is called a 'word' and each complete cycle of multiplexer operation, during which all channels are scanned once, results in the

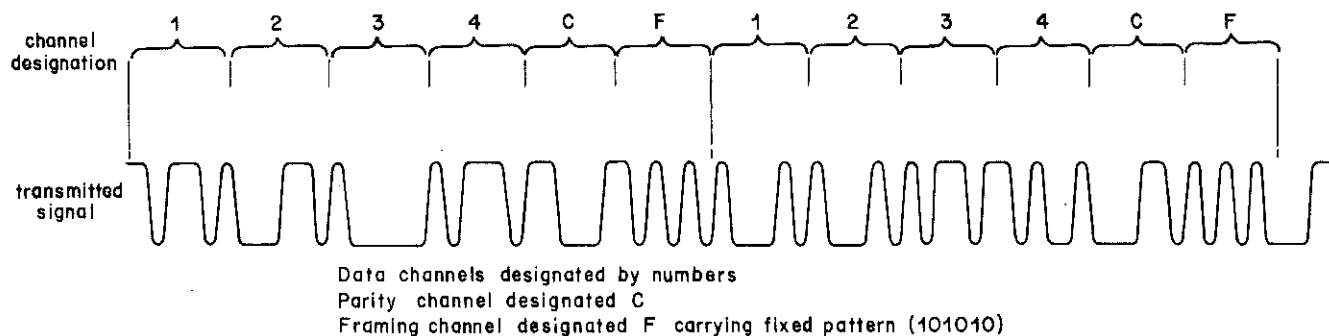


Fig. 2 - Simplified multiplex signal

transmission of a 'frame'. One frame will contain a word representing each sound channel and also some extra bits which convey control signals to synchronise the system and detect errors.

Fig. 2 shows, for example, two frames of a simplified multiplex signal having only four data channels with six bits per channel. The words representing sound signals 1 to 4 are inserted in the numbered time slots. In time slot F is a fixed pattern of bits known as a framing pattern. The receiving terminal equipment recognises this pattern and resets the demultiplexer to the start position ready for the next frame. Error detection bits, sometimes known as 'check bits', are inserted in the channel marked C. These bits are used to make an arithmetical check on the correctness of the data after transmission. In p.c.m. transmission of high quality sound signals the pulses are very rugged and in most circumstances will be correctly interpreted by the receiving equipment. When however the transmission link is subject to interference the pulses may be misread and output signal samples of random amplitude may occur. The resulting noises may be extremely disturbing and a method of detecting errors and concealing their effects is required. The concealment could in principle be done either digitally, i.e. before decoding, or by operating on the analogue signal after conversion.

3. Number of coded levels necessary

3.1. Use of compandors

The possibility of using a compandor to reduce the number of bits per word required to give a prescribed signal-to-quantising noise ratio has already been considered¹ with particular reference to the device of using non-uniformly spaced quantising steps in the coder and decoder. Subsequent tests⁴ have shown that the instantaneous companding thus effected introduces a form of distortion which makes it unsuitable for high-quality sound signals.

Syllabic compandors operating at audio frequencies can give satisfactory quality provided that their action is made frequency-dependent⁵ so that low-frequency components of the signal cannot vary the amplitude of high-frequency components of the noise; a compandor of this kind⁶ is incorporated in the 'Sound-in-Syncs' system referred to in Section 1.

For stereophony, however, an additional requirement must be imposed. Because the directional effects are

determined to a great extent during transients in the signal, it is essential that the overall gain of the compressor-expander combination in each channel should be constant under dynamic as well as static conditions; this requirement can be met, but only by considerable refinement in instrumentation. Thus, companding would introduce complex and costly analogue devices which would detract from the inherent stability, ruggedness and economy of a digital system. Only if the reduction in bit-rate achievable by a compandor (about 15% in the present case) were essential, could the extra cost and complexity be justified.

3.2. Desirable signal-to-noise ratio

Recent subjective tests⁷ have indicated that for a high-quality sound reproduction system in which the only audible noise is of the 'white' Gaussian form, the ratio of peak signal to peak weighted noise for the whole signal chain should be at least 60 dB. It is considered however, that not more than half the total noise power in the broadcasting chain should be contributed by the distribution network; for the latter, a signal-to-noise ratio of 63 dB should therefore be aimed at.

If, as was pointed out in an earlier report,¹ all mixing and routing operations in the programme network were carried out on the sound signals in their digital form, only one coding and decoding process would be necessary; a 12-bit p.c.m. code for which the theoretical signal-to-quantising noise, expressed in the same terms as before, is 65 dB, would then be adequate. However, a considerable period is likely to elapse between the introduction of digital distribution systems and the general adoption of digital techniques for the manipulation of signals; provision must therefore be made, in the first instance, for a number of p.c.m. links, each involving a coding and decoding process, to be connected in tandem. The BBC requirements are such that the maximum number of codecs (coder-decoder combinations) likely to be operated in tandem is four; to allow for this the required signal-to-noise ratio of each link must be increased to 69 dB and a 13-bit p.c.m. code, giving a signal-to-quantising noise ratio of 71 dB, is therefore sufficient.

3.3. Low-level signal distortion

Apart from quantising noise, non-linear distortion may be audible during very quiet passages of programme if

the signal excursion does not span a sufficiently large number of quantising steps. Experience with an experimental 11-bit p.c.m. system indicates that for the conditions existing in the V.H.F. radio service, where the dynamic range may be very wide and the average reproduced sound level high, this effect, known as 'granular distortion', needs to be taken into account.

Subjective tests carried out as part of this investigation and reported on separately⁸ show that 14 bits, representing 2^{14} coded levels, are necessary, in the absence of special measures, to eliminate audible granular distortion at very low signal levels. A reduction to 13 in the number of bits required was shown⁸ to be acceptable when an interpolating waveform, having a magnitude comparable with one quantising step, was added to the input signal, thereby effectively eliminating the granular distortion. When the interpolating waveform is used with 13 bit coding the signal-to-noise ratio is reduced to 69 dB which as indicated in Section 3.2 is sufficient. For the purpose of the present investigation however experimental equipment capable of conversion to 14-bit accuracy was developed.

4. Sampling frequency

For a high-quality music channel, a nominal upper frequency limit of 15 kHz should be aimed at i.e. the loss at this frequency relative to that at mid-band should not exceed 3 dB. Limitations of practical filters would then dictate a minimum sampling frequency of about 33 kHz.

For a p.c.m. system distributing stereophonic programmes to broadcasting transmitters operating on the Pilot Tone (Zenith — G.E.) system, there may be some advantage in sampling at 38 kHz — the frequency of the subcarrier used. The main components of the required stereo multiplex signal could then be generated at the transmitter input by switching, at 38 kHz, between the sample-held outputs of the two digital-to-analogue converters carrying the left-and-right-hand signals. This procedure would however involve a loss in channel capacity through the use of a higher sampling frequency than audio-bandwidth considerations would dictate.

For p.c.m. telephone systems, a sampling frequency of 8 kHz has been standardised, and it would therefore be advantageous to preserve compatibility with such systems by adopting a sampling frequency of 32 kHz for high-quality sound signals. To avoid unduly stringent filtering requirements, it might then be necessary to compromise slightly over the audio-frequency bandwidth transmitted, but the upper frequency limit could still be greater than 14 kHz.

For the experimental equipment used in the present study, the sampling frequency was chosen to be 38 kHz, thus covering the most difficult condition likely to be encountered.

5. Coding

In the Sound-in-Synchs system, referred to in Section 1, coding and decoding of the sound signal are performed by

10-bit ramp-counter converters⁹ in which 2^{10} coding levels are counted out in 25 μ s so that the counter frequency is 40 MHz. To count out 2^{14} levels in about the same time the counter frequency would have to be about 640 MHz, an impracticable figure at present.

In the experimental equipment built for the present study, in order to retain most of the virtues of ramp-counter coding⁹ without requiring very high counting speeds, the signal is first coarsely coded, using a high-slope voltage ramp, and the remainder is then finely coded, using a second ramp of lower slope. By this artifice the counting frequency is reduced to 15 MHz, and coding and decoding to 14-bit accuracy is accomplished to within 2 dB of the theoretical signal-to-noise ratio without great difficulty. Reducing the precision demanded to that provided by 13 bits, as indicated in Section 3.1, would make the requirements less stringent and the coder design easier; thus there is now no doubt of the technical feasibility of coding to the required accuracy. A more complete technical description of the coding and decoding equipment is given in a separate report.³

6. Multiplexing

In developing experimental multiplexing equipment an 8 Mb/s data rate was taken as the maximum likely to be required, thus presenting the most difficult multiplexing problem. The time slot into which each digit pulse must fit is thus $1/(8 \times 10^6)$ seconds, i.e. 125 ns wide. In order to avoid difficulties caused by propagation delays it is therefore desirable that the integrated circuits used to perform the multiplex switching should have propagation times which are short compared with 125 ns. Emitter-coupled logic circuits with propagation times of about 6 ns satisfy this requirement and current prices (5s. to 7s. per gate) are modest.

Methods employed in p.c.m. telephony are readily adapted to synchronise a multiplex receiver for high-quality digital sound signals. Extraction of timing information from the zero crossings of the data to generate a bit-rate clock by means of a high Q resonant circuit can be made to work satisfactorily even under very noisy transmission conditions by gating out those timing impulses from the data which are known to be widely in error.

Frame synchronisation by transmission of a regularly repeated framing pattern is satisfactory provided that means are adopted to prevent the receiving equipment being re-set by false framing patterns appearing in the data.

7. Transmission pulse shape and bandwidth

The merits of $(\cosine)^2$ pulses for distribution of digitally coded sound signals associated with television waveforms have been dealt with in an earlier report.¹⁰ Although in the television application it was necessary to crowd the pulses somewhat, the advantages of the $(\cosine)^2$ pulse in minimising intersymbol interference are fully realised only when the pulse spacing is equal to the half-height width. Assuming that this spacing were adopted,

the 8 Mb/s data rate of the experimental multiplex system could be readily transmitted in a base bandwidth of 8 MHz, a bandwidth available in S.H.F. links used for television. (Cosine)² pulses have been found immune to intersymbol interference errors even with pulse distortions for which K-ratings applied to this 8 MHz bandwidth, are as large as 10% – 15%.

8. Error detection and concealment

For the feasibility study an error detection system was fitted to the prototype demultiplexer which completely muted the programme output from all channels if the digit error rate became too high. This drastic course protected the listener from the ferocious noises which would otherwise occur. In an operational system, however, it would be preferable to use a more refined error concealment arrangement which, under adverse transmission conditions, would make the best of the situation, so that the programme need not be interrupted unnecessarily.

Investigation of error detection and concealment systems is being carried out as a separate matter and will be the subject of a later report.

9. Conclusions

Experience with experimental p.c.m. multiplex equipment confirms the feasibility of this system for the transmission of high-quality stereo sound signals.

Further development of error detection and suppression systems remains to be done to optimise the correction and/or concealment of transmission errors.

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